United States Patent [19]

Eaton et al.

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[54] **VOICE PROCESSING INTERFACE FOR A TELECONFERENCE SYSTEM**

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[51] Int. Cl.⁶ **H04M 3/56**

[51] Int. Cl.^{6} **H04M 3/56**

[52] U.S. Cl. **379/202**; 379/67; 379/88; 379/89; 379/62; 370/62; 370/110.1

[58] **Field of Search** 379/202, 201, 379/207, 67, 88, 89; 370/62, 110.1; 348/14, 16

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[57] **ABSTRACT**

A teleconferencing system having screened introductions, named introductions, roll call, talker identification, and subconferencing, scheduling, and recording options. A caller attempting to join a teleconference is prompted to enter identification information by generating corresponding DTMF signals. The caller's identification is used to index the caller's profile stored in memory. The profile includes a memory address pointer to the location at which data representative of the spoken name of the caller is stored. The caller's identification is also checked against a list specified by the meeting organizer and stored in memory to determine whether the caller is to be admitted to the teleconference. If the caller is to be admitted to the teleconference, an announcement is generated to the attendees of the teleconference using the caller's spoken name retrieved from memory. Likewise, when the attendee leaves the conference, the system detects the attendee disconnecting and retrieves the attendee's spoken name stored in memory in order to generate an announcement to the attendees: "XYZ has left the conference." The system is responsive to requests by attendees during the teleconference. The attendees communicate with the system by depressing buttons on their touch tone phones which generate DTMF signals. The system also has the capability of scheduling new conferences and automatically checking for any conflicts.

60 Claims, 12 Drawing Sheets
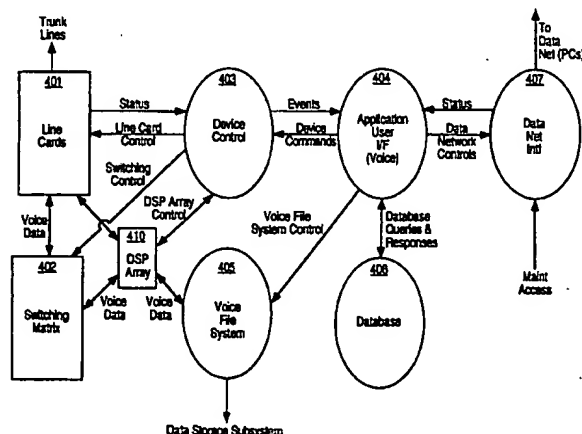
US-PAT-NO: 5483588

DOCUMENT-IDENTIFIER: US 5483588 A
See image for Certificate of Correction

TITLE: Voice processing interface for a
teleconference system

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Detailed Description Text - DETX (17):
   The voice file system 405, controlled by the user
application module 404,
stores and outputs voice data. More particularly, the voice
file system 405 is
coupled to the DSP array 410 and memory and enables real time
support of a
multi-port voice subsystem providing simultaneous playback
and record
operations as part of the automated teleconferencing services
described herein.
For example, when a caller wishes to join a teleconference,
the spoken name of
the caller is retrieved by the voice file system from the
data storage
subsystem and output to the DSP array 410. The DSP array 410
processes the
name and outputs the signals subsequently through the
switching matrix 402 to
line cards 401 for communication of a verbal announcement to
the conference
participants that the caller, identified by the spoken name,
is joining the
conference.

Current US Original Classification - CCOR (1):
   379/202.01

Current US Cross Reference Classification - CCXR (1):
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Current US Cross Reference Classification - CCXR (2):
    370/264

Current US Cross Reference Classification - CCXR (4):
    379/204.01